We claim:

- 1 1. A method, comprising the steps of:
- 2 (a) obtaining information relevant to the quality of service of voice calls being 3 transmitted from a first location to a second location via an IP network;
- 4 (b) calculating a parameter based on said information; and
- 5 (c) accepting a new call into the IP network in the case of said parameter not exceeding an upper threshold.
- 1 2. The method of claim 1 wherein said new call is accepted into the IP network at
- a reduced bandwidth in the case of said parameter exceeding a lower threshold.
- 1 3. The method of claim 1 where said new call is not accepted into the IP network
- 2 in the case of said parameter exceeding the upper threshold.
- 1 4. The method of claim 1 wherein the information obtained is a number of lost
- 2 packets, late packets and received packets (collectively defined as "sent" packets)
- 3 transmitted from said first location to said second location in the IP network.
- 1 5. The method of claim 1 wherein the information obtained is a delay of received
- 2 packets transmitted from said first location to said second location in the IP
- 3 network.
- 1 6. The method of claim 1 wherein the information obtained is a delay variation of
- 2 received packets transmitted from said first location to said second location in the IP
- 3 network.
- 1 7. The method of claim 1 wherein the information is obtained on a periodic basis.
- 1 8. The method of claim 1 wherein the information is obtained on an exception
- 2 basis using an immediate report.
- 1 9. The method of claim 1 wherein the parameter is identified as a packet lost
- 2 ratio (PLR).

10. The method of claim 9 wherein PLR is defined as

 $PLR = \frac{\text{(lost packets + late packets)}}{\text{(received packets + lost packets + late packets)}}$

- 1 11. The method of claim 2 wherein bandwidth is reduced for a newly accepted call
- 2 by selecting a first encoder to encode the new voice call information in a bandwidth
- 3 that is smaller than bandwidths of other calls accepted in the network that are
- 4 encoded by a second encoder.
- 1 12. The method of claim 2 wherein the bandwidth of a newly accepted call is
- reduced by increasing the packet size (voice sample) for said newly accepted voice
- 3 call.

1

- 1 13. The method of claim 2 wherein the bandwidth of a newly accepted call is
- 2 reduced by activating the characteristic of silence suppression for said newly
- 3 accepted voice call.
- 1 14. Apparatus comprising a gateway for interfacing voice call data from a public
- 2 switch telephone network to an internet protocol network; said gateway further
- 3 comprising:
- a first circuit for passing said voice call data to the internet protocol network;
- a second circuit for polling the internet protocol network about traffic
- 6 information transmitted therein; and
- 7 a third circuit for processing the polled information to determine whether the
- 8 voice call data is to be accepted by the internet protocol network.
- 1 15. The apparatus of claim 14 wherein said first circuit further comprises one or
- 2 more Ethernet cards that are connected to the internet protocol network.
- 1 16. The apparatus of claim 14 wherein said second circuit is at least one strongarm
- 2 card.
- 1 17. The apparatus of claim 16 wherein the strongarm card is connected to the
- 2 Ethernet card via a host CPU circuit.

- 1 18. The apparatus of claim 14 wherein the third circuit compares a parameter
- 2 based on the polled information to a plurality of thresholds.
- 1 19. The apparatus of claim 18 wherein the parameter is a packet loss ratio defined
- 2 as
- $PLR = \frac{\text{(lost packets + late packets)}}{\text{(received packets + lost packets + late packets)}}$
- 1 20. The apparatus of claim 19 wherein the third circuit compares the packet loss
- 2 ratio to a lower threshold and if the packet loss ratio is less than the lower threshold,
- a new voice call is accepted into the internet protocol network.
- 1 21. The apparatus of claim 19 wherein the third circuit compares the packet loss
- 2 ratio to the lower threshold and an upper threshold, and if lower threshold < packet
- loss ratio < upper threshold, a new voice call is accepted into the internet protocol
- 4 network at a reduced bandwidth.
- 1 22. The apparatus of claim 19 wherein the third circuit compares the packet loss
- 2 ratio to the upper threshold, and if the packet loss ratio is greater than the upper
- threshold, a new voice call is blocked from entering the internet protocol network.